FEEDBACK CANCELLATION

Problem

Feedback (FB) is a widespread problem with hearing aids (HA). The FB results in oscillations that are a source of annoyance to the user and to near-by individuals. The problem comes from the fact that there is a loop formed with the forward gain of the HA and the return through the HA vent or leakage around the device. When the total forward gain is greater then the attenuation of the return, path oscillation occurs. Strictly speaking there must also be a multiple of 360 degree phase shift around the loop for oscillation to occur. However, the delays in modern HA's are such that there is almost always a frequency where the phase requirement is satisfied.

Currently Available Solutions

The most common solutions to the FB problem are to either reduce the gain of the HA, such as turning down the volume control, or by adjusting the HA to fit tighter in the ear canal or to reduce the vent size. These are often unsatisfactory solutions since usually the forward gain is desired and a tighter fitting aid is less comfortable.

Other solutions have made use of electronics in the HA amplifier. One of the most attractive is the use of an internal FB path that is adjusted to match the external path. The internal path is subtracted from the input, resulting in a cancellation of the external path. Figure 1 shows a block diagram. The advantage of this design is that it cancels the FB without reducing the forward gain. A common approach is for the internal FB path to consist of a FIR filter whose coefficients are adjusted using a least mean squared (LMS) algorithm. This algorithm operates to reduce the power of signal at point E on the block diagram. If the input signal is NOT correlated with the signal coming to the mic via the FB path, then the algorithm works well.

The problem with the LMS algorithm is that there are many sounds where there is a high correlation of the input with the FB signal. If the input source is periodic, then the FB signal will correlate with the input. Musical inputs are a common example. Musical tones can last for a second or more which is much longer than the 2 to 12 ms that is typical of most HA FB loop delays. The result of the correlation is that the LMS algorithm adjusts the FIR filter to reduce the input signal, which in turn results in a misadjusted FIR filter. If the FIR filter becomes sufficiently misadjusted then a true FB oscillation will begin to build resulting in a very annoying artifact.

This problem with the LMS algorithm has been known for a long time and several ideas have been advanced to mitigate it. One idea has been to allow adjustment of the FIR filter only extremely slowly or only during initial fitting of the device. The weakness of this approach is that it means that there is poor or no compensation for changes in the FB that occur from common situations such as jaw motion or a telephone being brought near the ear. Another idea is to only allow the FIR a limited range of adjustment. This, however, also limits the range of correction that is possible. Another idea is to inject pseudo random noise into the output and look for that noise in the input. This works well if the noise is high enough but adding noise is not attractive to a HA user. Still another idea is to add a time varying delay in the forward path. This can break up the correlation of the FB signal from the input. The problem with this approach is that it requires the delay to change more rapidly than the FIR is corrected. In practical situations this rapid delay change results in a sound artifact that is undesirable.

Small Phase Shifting as a Measurement Means of the FB Path

My idea is to treat the condition where there is a highly correlated input in a unique manner. My algorithm first measures the correlation in the input signal. If there is very low correlation then the LMS algorithm can be used without a problem. However, if there is a high correlation then it is unclear whether the correlation is due to a highly correlated sound source, such as a musical tone or whether it is due to the FB path.

My idea is to make use of a small phase shift in the forward path as a means to measure the FB path. The phase shift would be small, typical about 20 degrees. If there is a net FB path, the phase shift will appear at the input after the loop delay time. A Phase Measurement circuit monitors the input and measures any phase changes that occur at the expected time. This measure is then used to adjust the FIR filter. This algorithm will be referred to as the Small Phase-Shift Detection (SPD) algorithm.

The SPD approach is an improvement over traditional LMS designs. Since the SPD algorithm depends on measuring a signal that is inserted (the phase change) it is not so dependent on the characteristic of the source. The SPD works well even when there is a highly correlated input, such as a pure tone. While the phase of the output is modified and hence is technically a type of distortion, it generally is unnoticeable to the user. The human ear has a low sensitivity to phase so the inserted phase shift is detectable by the algorithm circuits but it has a very tiny, usually undetectable, artifact to the listener.

Note that the SPD algorithm is distinct from the use of a varying delay in the forward path. The varying delay approach uses an LMS algorithm but with the time varying delay added to break up the correlation of the FB signal with the input. To accomplish this, the delay must change the phase of the signal by at least 180 degrees so that what was in phase becomes out of phase. Note also that this phase change must occur in a time shorter than the speed of the LMS adaptation. This typically means that either the adaptation must occur slower than desired or that the varying delay occurs so fast that it produces noticeable artifacts. The SPD is fundamentally different than the varying delay idea. The difference is that the SPD uses the delta phase not to break up the FB path but as a means to measure the FB path. Therefore, the phase changes of the SPD algorithm are much smaller than with the varying delay approach. The phase shift used in the SPD algorithm is in the range of 10 to 60 degrees. In the current implementation the phase shift is about 20 degrees.

Detailed Description of Small Phase Shift Detection Algorithm

Figure 2 shows a block diagram of the Small Phase-Shift Detection (SPD) algorithm. There is an FIR filter that functions to cancel the external path. This filter can be implemented in a manner similar to the LMS designs. There is a Correlation Detector that functions to determine when to use LMS and when to apply SPD adaptation. There is a Phase Shifter block that performs the shift on a band of frequencies when the correlation is detected. Lastly there is a Phase Measurement block that functions to detect any phase changes in the input do to the phase shifter. These blocks are described in more detail below.

The DSP processor that is the preferred platform is the Toccata DSP system from dspfactory, Ltd, 611 Kumpf Drive, Unit 200, Waterloo, Ontario, N2V1K8, Canada. This DSP uses a Weighted Overlap-Add filter bank coprocessor that functions to divide the incoming signal into frequency sub bands. This processor is very efficient for algorithms, such as the SPD, that work on distinct frequency bands. Other DSP's could be used to implement the blocks of SPD algorithm and the following descriptions include implementation with both general DSP's and the Toccata DSP.

Correlation Detector

The correlation detector concept is to compare the incoming signal with a delayed version of the input. When the average of product of the input with the delayed input is high then we say there is a high correlation. The delay of the correlation that is particularly important is the delay that approximates the total delay through the HA FB loop. Typically this is about 6ms through the forward path plus about 2 ms through the FB path. The correlation is calculated as follows:

$$C(m) = \text{Ave}_n[\text{In}(n\tilde{)}\ \ \text{In}(n-m)]$$

Where C(m) is the correlation at m delay
Ave_n is an averaging over the time variable
n, and In(n) is the input signal.

The correlation for the Toccata DSP uses a similar calculation but performs it on the output of the filter banks.

$$C_i(m) = \text{Ave}_n[E_i\,(n\tilde{)}\ \ E_i{}^*(n-m)]$$

Where C(m) is the correlation at m delay
Ave_n is an averaging over the time variable
n, and E_i (n) is the signal from the i^{th} filter
band

The Toccata provides efficient band filtering so that there is a correlation function for each band of interest. Since the output of the filter banks is a complex number, the product in the above formula uses the complex conjugate for the second term (i.e. E*(n-m)). The averaging used with the current implementation on Toccata calculates the standard deviation of the C(m) for 16 input samples (n's). This value is then compared to the mean value of C(m) for the same 16 samples. If the standard deviation is greater than 0.7 of the mean then the correlation is determined to be "low". Experiments indicate that a deviation to mean ratio in the range of .25 to 1.0 is the threshold to use.

If the correlation is low then the input is relatively "random". There is no FB oscillation and no periodic signal source. In this case it is generally safe to use the LMS algorithm if the convergence speed is relatively slow. Note that it is not necessary, in this case, to converge rapidly since there is no actual FB oscillation.

If the correlation is high it means that there is periodic or nearly periodic input. This can be the result of either a true periodic sound source or it could also result from FB oscillation. The correlation detector will show a high level in both cases but not distinguish between the two. The LMS algorithm is NOT a good choice to use here. If the situation is FB oscillation, the convergence speed should be fast. However, for the case of a periodic input, the convergence should very slow. Therefore, when there is high correlation we switch to using the SPD algorithm.

Phase-Shifter Block

The Phase-Shifter Block functions to add a distinct phase change to the output of the aid that will allow later detection at the input. The phase is small enough to have insignificant perception to the HA user. Figure 3 shows some possible phase shapes. The phase pattern currently used is the "moderate duration" ramped pulse shown in Figure 3 -B. The ramping of the pulse reduces perceptual artifacts. The pulse shape allows for a simpler detection of the phase when it appears at the input.

The Phase-Shifter Block could be implemented using an all-pass filter or other phase shifting means. With the Toccata DSP, the phase shift is done by multiplying the filter bank output by the complex number [COS(a) +j SIN(a)] where a = phase shift angle.

Phase-Measurement Block

The Phase-Measurement Block functions to measure the phase at the input. The measurement timing is synchronized with the phase change inserted on the output. The phase at the input is measured after a delay about equal to the loop delay. If there is no input phase change in response to the output change then there is no FB. If there is an input phase change, the direction of it indicates how the FIR filter coefficients should be changed to minimize the net FB path. Figure 3 shows the timing of the measurement.

One means of implementing the phase measurement is to use a phase locked loop. This circuit would synchronize an oscillator with the input and then any input phase change would be measurable as a phase differential between the input and the oscillator. The implementation used with the Toccata DSP uses the Correlation Detector circuit. The Correlation Detector compares the input vs. a delayed version of the input. If there is a phase change at the input it can be measured as a shift relative to the delayed input. Figure 4 shows details.

FIR Filter and Coefficient Update

An FIR filter is used to provide the internal FB path. Such FIR filters are common in adaptive algorithms. When used with the Toccata DSP there is, however, an unusual aspect regarding the filter coefficient updates. The Toccata DSP, as previously stated, works on narrow frequency bands. Thus, the result of the phase measurement is an indication of the desired FB path change but only on a particular frequency band of the FB path. Therefore, the updates to the coefficients are organized not as individual taps of the filter but as groups that affect a particular narrow frequency band.

For example, consider a 32 tap FIR filter, sampled at 16 kHz. See Figure 5. The taps can be organized into 16 filter bands centered at 0, 0.5, 1.0 ...7.0, 7.5 kHz. For each filter there are two sets of coefficients that differ by 90 degrees. For the above example at 4 kHz, one set of coefficients is:

$$a(n) = COS\left(n \bullet 360\,\text{deg} \bullet \frac{4kHz}{16kHz}\right) \quad \text{for } n = 0,1,\ldots 31 \qquad \text{[equation 1]}$$

The other set of coefficients for 4 kHz is:

$$b(n) = SIN\left(n \bullet 360\,\text{deg} \bullet \frac{4kHz}{16kHz}\right) \quad \text{for } n = 0,1,\ldots 31 \qquad \text{[equation 2]}$$

These coefficients are fixed. The frequency bands are formed from these coefficients as follows:

$$\mathrm{f1cos} = \sum_n a(n) \bullet x(n) \quad \text{where } x(n) = \text{input samples}$$

and

$$\mathrm{f1sin} = \sum_n b(n) \bullet x(n) \quad \text{where } x(n) = \text{input samples}$$

The final output of the filter is formed by combining the frequency bands as follows (see Figure 5):

$$\text{Output} = \sum_m \mathrm{Cf}m\mathrm{cos} \bullet \mathrm{f}m\mathrm{cos} + \mathrm{Cf}m\mathrm{sin} \bullet \mathrm{f}m\mathrm{sin}$$

The "C" coefficients are adjusted in the adaptation process to change the FIR filter shape. There is the same number of independent variables when the coefficients are grouped by frequency bands as when they are treated individually. The advantage of frequency grouping is that updating Cfm affects only one frequency band.

The above FIR filter design is computationally efficient for adaptation changes since the "C" coefficients are directly related to the frequency band being changed. However, this filter architecture is not computationally efficient for the basic filter computation since the total number of coefficient multiplications is high. The basic filter computation occurs at the input sampling rate, while the adaptation occurs at a slower rate. Therefore it is generally more important for the basic filter computation to be the most efficient. Because of this, the actual filter architecture implemented on the Toccata DSP is the standard form shown in Figure 6 and as follows:

$$\text{Output} = \sum_n C(n) \bullet x(n) \qquad \text{where } x(n) = \text{input samples}$$
$$\text{and } C(n) = \text{coefficients}$$

This allows the basic filter computation to be done efficiently. The adaptation, however, is more complicated with this architecture. The coefficients must be updated in a manner that changes the filter output at one frequency band. This is done by adding a fraction of the appropriate band coefficients a(n) or b(n) (equations 1 or 2) from above. Specifically:

$$\text{Cnew}(n) = \text{Cold}(n) + \lambda \bullet a(n) \qquad \text{or}$$

$$\text{Cnew}(n) = \text{Cold}(n) + \lambda \bullet b(n) \qquad \text{where } C(n) = \text{coefficients}$$

$a(n) =$ cosine coefficients for particular freq band
$b(n) =$ sine coefficients for particular freq band
and λ = adaptation direction and rate

Operation of SPD system with Toccata DSP

In order to clarify the operation of the SPD system, we will describe the operation as implemented on the Toccata DSP. We will describe the performance under three important conditions: (a) FB that is sub-oscillator but greater than 0, with a random noise input. (b) Zero net FB with a 4 kHz sinusoidal input and (c) FB sufficient to cause strong oscillation with a low-level signal input. The first case simulates most normal use with speech or uncorrelated inputs where some correction of the net FB is desirable but can be done slowly. The second case (b) simulates many musical conditions where no correction of the FIR is desired. The last case (c) simulates the condition of FB oscillation where speedy correction of the FIR is desired.

The Correlation Detector performs the first operation. The detector discriminates between the conditions of random noise (a) and the other conditions. If the detector determines that the input in non-correlated (a) then the FIR filter is updated using an LMS algorithm. The LMS algorithm has been shown to work well for these conditions.

If the Correlation Detector determines that there is a high correlation at the input then the source is either periodic (condition b), or there is FB oscillation (condition c). In either case, the Phase-Shifter is started. In this example, the high correlation occurs in the 4 kHz region so the output phase is changed over the 4 kHz band. The phase at the output is altered as shown in

Figure 3-B. Since the phase change is only 20 degrees the frequency response perturbations at the band edges are insignificant.

Next the Phase Measurement Block comes into play. Suppose for this example that the total loop delay is 8 ms. The phase measurement would then begin 8 ms after the output phase-shift began. See Figure 3-B. For the condition (b), where the source is a sinewave, there will be no phase detected at the input because there is no net FB path in that condition. Since there is no phase shift detected, there is no alteration to the FIR coefficients. This is the desired response, since the net FB is already zero.

For the condition (c), where there is FB oscillation, there will be a phase change detected at the input. This is because the input is largely due to the FB signal, which has been phase shifted. In the Phase Measurement Block, as implemented on the Toccata DSP, the phase is a complex number, (i.e. it is a vector). Hence the phase measurement result has an angle between 0 to 360 degrees. In the current implementation, the angle is approximated by 0, 90, 180 or 270 degrees, which ever is the closest. The angle chosen is used to select which of the four directions to correct the FIR coefficients. For example, the 0 degree angle would add the COS coefficients, the 90 degree angle would add the SIN coefficients. The 180 degree angle would subtract the COS coefficients and the 270 degree angle would subtract the SIN coefficients. Which angle would control which coefficient set is determined experimentally for each frequency band. The coefficients updated are the 4 kHz set since it is in the 4 kHz band that the phase shift was applied and measured. The magnitude of the coefficient correction that is applied is determined by the desired speed of convergence.

In this example, the cycle of correlation detection through coefficient update is from about 20 to 40 ms. After one cycle is completed a new cycle is started. The adaptation runs continuously, allowing the system to respond to changes that occur in the external FB path such as when objects are moved close to the ear or the fit of the aid in the ear canal changes.

Variations and Improvements on the SPD Algorithm

Several variations on the basic SPD algorithm are possible that may offer improved performance under certain conditions:

1) One possibility is to use differing levels of phase shift depending on the amount of correlation detected at the input. Less phase shift is desirable for highly correlated inputs such as pure tones. The smaller phase shift is desirable for the pure tone inputs since perceptual artifacts are more noticeable with these inputs. With these inputs, the inserted phase shift is also more readily measured at the input so small phase shifts still work fine. On the other hand, for sounds that are only somewhat correlated, larger phase shifts are less noticeable but desired to improve detection.

2) Another improvement may be to set up a hold-off of the LMS for a certain time after a tone is detected in any band. The idea is that, if there is a clear indication that there is a tonal input in any band, the input is likely to be musical even if the correlation detector does not detect it in all bands. This technique has been helpful in preventing complex musical signals from being treated by LMS.